JC07 Rec'd PCT/PTO 06 APR 2001

| FORM PTO-1390 (REV. 5-93) U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE | ATTORNEY'S DOCKET NUMBER **10191/1822** |
|---|---|
| **TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371** | U.S. APPLICATION NO. (If known, see 37 CFR 1.5) |

| INTERNATIONAL APPLICATION NO. | INTERNATIONAL FILING DATE | PRIORITY DATE CLAIMED: |
|---|---|---|
| **PCT/DE99/02633** | **21 August 1999 (21.08.99)** | **6 October 1998 (06.10.98)** |

TITLE OF INVENTION
METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER

APPLICANT(S) FOR DO/EO/US
**Torsten PRANGE, Andreas ENGELSBERG, Christian MITTENDORF, and Torsten MLASKO**

Applicants herewith submit to the United States Designated/Elected Office (DO/EO/US) the following items and other information.

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) immediately rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).
4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.
5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2))
   a. ☐ is transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☒ has been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US)
6. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3))
   a ☐ are transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ have. been transmitted by the International Bureau.
   c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   d. ☒ have not been made and will not be made.
8. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
9. ☒ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)) (unsigned).
10. ☒ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

**Items 11. to 16. below concern other document(s) or information included:**

11. ☒ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
12. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
13. ☒ A **FIRST** preliminary amendment.
14. ☒ A substitute specification.
15. ☐ A change of power of attorney and/or address letter.
16. ☒ Other items or information: International Search Report (translated), Preliminary Examination Report and PCT/RO/101.

EXPRESS MAIL NO.:

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| U.S. APPLICATION NO. (if known, see 37 C.F.R.1.5 09/807015 | INTERNATIONAL APPLICATION NO PCT/DE99/02633 | ATTORNEY'S DOCKET NUMBER 10191/1822 |
|---|---|---|

17. ☒ The following fees are submitted:

**Basic National Fee (37 CFR 1.492(a)(1)-(5)):**

Search Report has been prepared by the EUROPEAN PATENT OFFICE or JPO .......... $860.00

International preliminary examination fee paid to USPTO (37 CFR 1.482) ..... $690.00

No international preliminary examination fee paid to USPTO (37 CFR 1.482) but international search fee paid to USPTO (37 CFR 1.445(a)(2)) .......... $710.00

Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO .......... $1,000.00

International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(2)-(4) .......... $100.00

| | | | | CALCULATIONS | PTO USE ONLY |
|---|---|---|---|---|---|
| **ENTER APPROPRIATE BASIC FEE AMOUNT =** | | | | $ 860 | |
| Surcharge of $130.00 for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(e)). | | | | $ | |
| Claims | Number Filed | Number Extra | Rate | | |
| Total Claims | 14 - 20 = | 0 | X $18.00 | $ 0 | |
| Independent Claims | 2 - 3 = | 0 | X $80.00 | $ 0 | |
| Multiple dependent claim(s) (if applicable) | | | + $270.00 | $ | |
| **TOTAL OF ABOVE CALCULATIONS =** | | | | $ 860 | |
| Reduction by ½ for filing by small entity, if applicable. Verified Small Entity statement must also be filed. (Note 37 CFR 1.9, 1.27, 1.28). | | | | $ | |
| **SUBTOTAL =** | | | | $ 860 | |
| Processing fee of $130.00 for furnishing the English translation later the ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(f)). + | | | | $ | |
| **TOTAL NATIONAL FEE =** | | | | $ 860 | |
| Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31). $40.00 per property + | | | | $ | |
| **TOTAL FEES ENCLOSED =** | | | | $ 860 | |
| | | | | Amount to be: refunded | $ |
| | | | | charged | $ |

a. ☐ A check in the amount of $__________ to cover the above fees is enclosed.

b. ☒ Please charge my Deposit Account No. 11-0600 in the amount of **$860.00** to cover the above fees. A duplicate copy of this sheet is enclosed.

c. ☒ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. 11-0600. A duplicate copy of this sheet is enclosed.

**NOTE:** Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

By: [illegible] (Reg. No. 41,172)

Richard L. Mayer

SIGNATURE

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4/6/01

DATE

26646

PATENT TRADEMARK OFFICE

[10191/1822]

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

| | | |
|---|---|---|
| Applicant(s) | : | PRANGE et al. |
| Serial No. | : | To Be Assigned |
| Filed | : | Herewith |
| For | : | METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER |
| Art Unit | : | To Be Assigned |
| Examiner | : | To Be Assigned |

Assistant Commissioner
for Patents
Washington, D.C. 20231
Box Patent Application



**PRELIMINARY AMENDMENT AND**
**37 C.F.R. § 1.125 SUBSTITUTE SPECIFICATION STATEMENT**

SIR:

Please amend the above-identified application before examination, as set forth below.

**IN THE SPECIFICATION AND ABSTRACT:**

In accordance with 37 C.F.R. § 1.121(b)(3), a Substitute Specification (including the Abstract, but without claims) accompanies this response. It is respectfully requested that the Substitute Specification (including Abstract) be entered to replace the Specification of record.

**IN THE CLAIMS:**

On the first page of the claims, first line, change "Patent Claims" to:
--What Is Claimed Is:--.

Please cancel original claims 1 to12, without prejudice, in the underlying PCT Application No. PCT/DE99/02633, and cancel substitute claims 1-10.

Please add the following new claims:

11. (New) A method for one of coding and decoding speech signal sampled values, comprising the steps of:

quantizing values previously obtained by an analysis from the speech signal sampled values and used for a generation of speech signal parameters before being stored in code books/code tables, the quantizing occurring to a word length that results in no noticeable losses in speech quality;

storing in the code books/code tables the values previously obtained by the analysis from the speech signal sampled values and used for the generation of speech signal parameters;

scaling the values of each code book/code table such that an available range of values is exploited as completely as possible, the step of scaling including the steps of:

determining a maximum of a positive value and a negative value of each code book/code table,

if the available range of values is exceeded, performing a multiplication of the values of each code book/code table by a first factor smaller than one, and

repeating the multiplication until all elements are located in the available range of values; and

causing a number of repeated multiplications to be used as a scaling factor for all code book/code table entries.

12. (New) The method according to claim 11, wherein:

the method is performed in accordance with a method of analysis through synthesis.

13. (New) The method according to claim 11, wherein:

the noticeable losses in speech quality are determined through a hearing test.

14. (New) The method according to claim 11, wherein:

the first factor is 0.5.

15. (New) The method according to claim 11, further comprising the step of:

determining word lengths of the values stored in the code books/code tables through hearing tests.

16. (New) The method according to claim 11, further comprising the step of:

scaling the code book/code table entries to bits of a required value range.

17. (New) The method according to claim 16, further comprising the step of:

for a finally valid quantization, performing a rounding and a subsequent truncation of decimal places.

18. (New) The method according to claim 11, wherein:

the word length is 16 bits.

19. (New) The method according to claim 11, further comprising the step of:

causing a processing of the code book/code table entries to occur in accordance with a digital signal processing in a whole-number format.

20. (New) The method according to claim 11, wherein:

for a HXVC (Harmonic Vector Excitation Coding) speech coder/speech decoder, LPC coefficients, spectral envelopes of a speech signal, and unvoiced segments of the speech signal are stored in quantized form in corresponding ones of the code books/tables.

21. (New) The method according to claim 11, wherein:

for a CELP (Code Excited Linear Prediction) speech coder/decoder, values for LSP (Line Spectral Pairs) VQ vector quantization code book/table entries, as well as those of gain VQ table entries, are stored in quantized form.

22. (New) An apparatus corresponding to one of a coder and a decoder for processing speech signal sampled values in accordance with a method of analysis through synthesis, comprising:

an arrangement for storing in quantized form values contained in code books/code tables for a generation of speech signal parameters;

an arrangement for selecting a word length such that no noticeable losses in speech quality occur;

an arrangement for quantizing the values contained in the code books/code tables to the word length that results in no noticeable losses in speech quality;

an arrangement for scaling the values of each code book/code table such that an available range of values can be exploited as completely as possible;

an arrangement for determining a maximum of positive values and negative values of each code book/code table, and for multiplying the values of each code book/code table by a first factor less than one if the available range of values is exceeded; and

an arrangement for, if a multiplication of the values of the code books/code tables lies outside the available range of values, performing a repeated multiplication until all elements are located in the available range of values, and for providing a number of repeated multiplications as a scaling factor.

23. (New) The apparatus according to claim 22, wherein:

the noticeable losses in speech quality are determined through a hearing test.

24. (New) The apparatus according to claim 22, wherein:

the first factor is 0.5.

## Remarks

This Preliminary Amendment cancels original claims 1 to 12 without prejudice, in the underlying PCT Application No. PCT/DE99/02633, and cancels substitute claims 1-10, without prejudice. The Preliminary Amendment also adds new claims 11-24. The new claims conform the claims to U.S. Patent and Trademark Office rules and do not add new matter to the application.

In accordance with 37 C.F.R. § 1.121(b)(3), the Substitute Specification (including the Abstract, but without the claims) contains no new matter. The amendments reflected in the Substitute Specification (including Abstract) are to conform the Specification and Abstract to U.S. Patent and Trademark Office rules or to correct informalities. As required by 37 C.F.R. § 1.121(b)(3)(iii) and § 1.125(b)(2), a Marked Up Version Of The Substitute Specification comparing the Specification of record and the Substitute Specification also accompanies this Preliminary Amendment. Approval and entry of the Substitute Specification (including Abstract) are respectfully requested.

The underlying PCT Application No. PCT/DE99/02633 includes an International Search Report, dated February 23, 2000, and an International Preliminary Examination Report dated January 15, 20001, copies of which are submitted herewith

Applicants assert that the subject matter of the present application is new, non-obvious, and useful. Prompt consideration and allowance of the application are respectfully requested.

Respectfully Submitted,

KENYON & KENYON

By: Leo Mangut (Reg. No. 41,172)

Dated: 4/6/01

By: Richard L. Mayer

Richard L. Mayer
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[10191/1822]

# METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER

## Field Of The Invention

The present invention relates to a method for coding or decoding speech signal sampled values.

## Background Information

In the standard for coding audiovisual objects according to MPEG-4, in ISO/IEC 14496-3 FCD, Subpart 2, parametric coders are specified, in particular the HVXC (Harmonic Vector Excitation Coding) coder, for coding speech at extremely low bitrates. In order to generate the LPC coefficients, the spectral envelopes of the speech signal, and the unvoiced segments, this standard contains a plurality of tables that are present in floating-point format.

In Subpart 3 of this standard, the CELP (Code Excited Linear Prediction) coder for coding speech at medium to low bitrates is described. For generating the LPC coefficients and the gain values, this standard contains a plurality of tables that are present in floating-point format.

For coding such speech signals, the method of "analysis through synthesis" is often used (ANT Nachrichtentechnische Berichte, Heft 5, Nov. 1988, pages 93 to 105). In the mentioned speech coding methods, values are stored in code books, i.e., in the tables, the values being used for the generation of the signal parameters and thus for the coefficients of the speech synthesis filter. The values stored in the code books are read out via an index control unit.

## Summary Of The Invention

Through the quantization of the values in the code books, the existing data are limited in their precision (quantization) so that the code book entries can be represented with a finite word

length. In this way, their transfer to digital signal processors with whole-number arithmetic can take place without infringing the quality demands prescribed by standards, in particular according to ISO/IEC 14496-3. In contrast to the present invention, in the mentioned working versions of the standards the values for the code books are present in unquantized form, in floating-point format, and can be processed directly only using very expensive and memory-intensive methods. Despite the limitation of precision of the table values, in the present invention an equal subjective quality is to be achieved after the speech decoding. Using the measures of the present invention, a simple transfer – conforming to standards – of the code to various computing platforms is possible without influencing the subjective quality of the coder. Since reduced word lengths are used, a considerable savings of memory capacity, in particular in the form of ROMs, is possible. The present invention can be used with various speech signal coding methods, for example for HVXC coders/decoders or CELP coders/decoders.

## Brief Description Of The Drawings

Figure 1 shows a simplified block switching diagram of an HVXC speech decoder.

Figure 2 shows a simplified block switching diagram of a CELP speech decoder.

## Detailed Description

Before discussing the actual quantization, a speech decoder is first presented in which the inventive quantization is used.

In the HVXC speech decoder according to Figure 1, the transmitted speech parameters, namely the LPC parameters, the voiced/unvoiced decision of the encoder, and the excitation parameters, which are contained in a transmission frame of 20 ms duration, are read out from the bitstream and are supplied as input signals to inputs 1, 2, and 3. The LPC parameters contain indices from which inverse LSP vector quantizer 16 regenerates the LSP (Line Spectral Pairs) parameters. For this purpose, LSP code books 4 (CbLsp) and 5 (CbLsp4) are indexed with the LPC parameters, and the LSP parameters are read out. Dependent on the

voiced/unvoiced decision of this frame, if necessary interpolation – module 6 – takes place between the LSP parameters of the past and current frame, achieving an updating of these values in a raster of 2.5 ms. Subsequently, conversion takes place into LPC parameters, which enter as coefficients into the LPC synthesis filter – modules 7 and 8.

Parallel to this calculation, and as a function of the voiced/unvoiced decision, the vectors for the spectral envelope (voiced frame), AM code books 9 (CbAm) and 10 (CbAm4), or the vectors for the stochastic excitation signal (unvoiced frame, CELP code books 11 (CbCelp) and 12 (CbCelp4)) are read. The regeneration of the spectral envelopes and of the excitation signal takes place using the inverse vector quantizers 13 and 14. After the harmonic synthesis (voiced) – module 15 – the filtering of the speech data takes place in the LPC synthesis filter. The output data from the voiced - module 7 - and from the unvoiced - module 8 - synthesis filter are subsequently added, yielding the reconstructed speech signal for a frame of 20 ms.

Because, as explained above, values for the code books in floating-point form are not suitable for fixed-point DSPs, because the required word lengths would be too large (memory requirement, internal word lengths and arithmetic, ROM), the conversion of the table values for the code books that were previously obtained by analysis from the speech signal sampled values takes place in a quantized form, with resulting equivalent speech quality. The word lengths required for this for the individual table values are determined in various hearing tests.

The quantization takes place to a word length that is determined in various tests. In the following representation, this word length is designated in general as *wordlength*. This size is expressed in bits. A signed whole number having *wordlength* bits includes a value range from $-2^{wordlength-1}$ to $2^{wordlength-1}-1$. The quantization of the code books in this context takes place in the manner shown below. The beginning point is represented by the code books defined in the "Study on ISO/IEC 14496-3 FCD, Subpart 3." For this document, the code book *cb* is defined as follows: $cb = \{a0, a1, \ldots, an, \ldots, am\}$ with $0 \leq n \leq m$ and $an \in R$. For the quantization of the individual elements, the following steps are required:

1.) Determination of the value range of the code books

In order to obtain a well-matched quantization, the elements of each code book are scaled in such a manner that the available value range is exploited as completely as possible. For this purpose, the value range of the elements is located between

$$\frac{-2^{wordlength-1}}{2^{wordlength-1}} = -1 \quad \textbf{and} \quad \frac{2^{wordlength-1}-1}{2^{wordlength-1}} = 1-2^{-(wordlength-1)}$$

In order to achieve this, the maximum of the positive and of the negative elements (*max_pos* or *max_neg*) of each code book is determined. These result from

$max\_pos = \max\left(\{a_n \in cb | a_n \geq 0\}\right)$ or $max\_neg = \min\left(\{a_n \in cb | a_n \geq 0\}\right)$, with $0 \leq n \leq m$.

As a function of the magnitude of *max_pos* or *max_neg*, the following steps result:

$max\_pos > (1 - 2^{-(wordlength-1)})$ or $max\_neg < -1$

*max_pos* and *max_neg* are multiplied by ½. If the result still satisfies the condition set under (a), then the process is repeated until the condition no longer holds. The number of multiplications by ½ is counted and is stored in the variables *scale*.

$max\_pos \leq (1 - 2^{-(wordlength-1)})$ or $max\_neg \geq -1$

*max_pos* and *max_neg* are multiplied by 2. If the result still satisfies the condition set under (b), then the process is repeated until the condition no longer holds. The number of multiplications by 2 is counted and is stored in the variables *scale*.

2.) Scaling of the elements of *cb* to the range between -1 and $(1 - 2^{-(wordlength-1)})$.

As a function of the decision made under 1.), the scaling of all code book entries to the cited range takes place:

$$b_n = \frac{1}{2^{scale}} a_n \forall a_n \in cb \text{ with } 0 \le n \le m$$

$$b_n = 2^{scale} a_n \forall a_n \in cb \text{ with } 0 \le n \le m.$$

After this step, the entries of each code book are located in the following range of values: $-1 \le b_n \le (1 - 2^{-(wordlength-1)})$, with $0 \le n \le m$.

3.) Scaling to *wordlength* bits

For the scaling to the required value range, multiplication by $2^{wordlength-1}$ takes place. In this way, the values of code books $c^n$ are located in the range between $-2^{wordlength-1}$ and $2^{wordlength-1}-1$.

4.) Rounding

Before the decimal places are truncated, rounding of the determined entries takes place. For this purpose, depending on the sign + 0.5 or – 0.5 is added. This takes place in the following form:

$$c_n \ge 0 : d_n = c_n + 0.5$$
$$c_n < 0 : d_n = c_n - 0.5$$

Here care is to be taken not to exceed the maximum permissible value range. This is located in the range as indicated under 2.).

5.) Truncation of the decimal places

The final quantization takes place through the truncation of the decimal places. The quantized values are obtained in this way.

Trials have shown that with the setting of the variables *wordlength* at 16, a speech quality indistinguishable from the original is obtained.

A further construction of the present invention is explained in connection with Figure 2.

There, the block switching diagram of a CELP decoder is shown. First, the elements for decoding a frame are read from a transmitted bitstream, as before. These include the LPC indices, the excitation parameters (lag and shape index), and the amplitude indices (gain indices). These parameters (elements) are supplied to decoder inputs 17 to 21. The excitation parameters are made up of the parameters for adaptive code book (lag) 22 for the generation of periodic signal components (voiced) and the parameters for fixed code books (shape index) 23a ... 23n.

The entries of fixed code books 23a ... 23n and of adaptive code book 22 are each multiplied by a scaling factor (gain) via gain decoder 24. This scaling factor is reconstructed with the aid of the gain indices present at the input 21 and the gain VQ (vector quantization) tables stored in code books 25. The finally valid excitation vector is composed from the sum of the fixed and the adaptive code book vector.

With the use of vector quantizer VQ, the LPC indices represent the vector-quantized LSP (Line Spectral Pairs) parameters. The vectors of the first and second stage of the inverse vector quantization of the LSP parameters are obtained by reading out the LSP-VQ table values, which are stored in code books 26. The finally valid reconstruction of the LPC parameters takes place in LPC parameter decoder 27. Inside each frame, for each subframe interpolation - module 28 - takes place between the LSP parameters of the past and of the current frame. The LSP parameters, converted into LPC parameters, enter into LPC synthesis filter 29 as coefficients. The reconstruction of the speech data takes place there through filtering of the excitation signal. In order to improve the speech quality, the reconstructed speech signal can be additionally filtered in a post-filter 30.

The LSP VQ table values, as well as the gain VQ table values for code books 25 and 26, which were previously obtained by analysis from the speech signal sampled values, are normally present in a floating-point representation, which, as explained above, is not suitable for a fixed-point DSP processing. For the same reasons as in the case of the HVXC decoder (Figure 1), a conversion of the table values into a quantized form takes place. The method steps in this quantization, such as in particular the determination of the value range for the code books, takes place as in the previously explained quantization.

The above exemplary embodiments of the present invention have been explained on the basis of speech decoders. Of course, the present invention can also be used in corresponding coders (encoders) that use code books. There as well, the code book entries can be previously quantized for the preparation of speech signals for transmission. Examples of such encoders whose code book entries can be previously quantized described in European Published Patent Application No. 0545 386, U.S. Patent No. 5,208,862, U.S. Patent No. 5,487,128, U.S. Patent No. 5,199,076, or U.S. Patent No. 5,261,027.

## Abstract Of The Disclosure

For the coding or decoding of speech signal sampled values, the values contained in the code books/code tables for the generation of the speech signal parameters are stored in quantized form. The processing can be carried out using processors with whole-number processing, without deterioration of the speech quality.

09/807015
JC02 Rec'd PCT/PTO 06 APR 2001
[10191/1822]

# METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER

[Prior art] Field Of The Invention

The present invention [is based on] relates to a method for coding or decoding speech signal sampled values.

Background Information

In the standard for coding audiovisual objects according to MPEG-4, in ISO/IEC 14496-3 FCD, Subpart 2, parametric coders are specified, in particular the HVXC (Harmonic Vector Excitation Coding) coder, for coding speech at extremely low bitrates. In order to generate the LPC coefficients, the spectral envelopes of the speech signal, and the unvoiced segments, this standard contains a plurality of tables that are present in floating-point format.

In Subpart 3 of this standard, the CELP (Code Excited Linear Prediction) coder for coding speech at medium to low bitrates is described. For generating the LPC coefficients and the gain values, this standard contains a plurality of tables that are present in floating-point format.

For coding such speech signals, the method of "analysis through synthesis" is often used (ANT Nachrichtentechnische Berichte, Heft 5, Nov. 1988, pages 93 to 105). In the mentioned speech coding methods, values are stored in code books, i.e., in the tables, [said] the values being used for the generation of the signal parameters and thus for the coefficients of the speech synthesis filter. The values stored in the code books are read out via an index control unit.

[Advantages of the] Summary Of The Invention

Through [the measures of Claim 1, i.e., in particular through] the quantization of the values in the code books, the existing data are limited in their precision (quantization) so that the code book entries can be represented with a finite word length. In this way, their transfer to digital signal processors with whole-number arithmetic can take place without infringing the quality demands prescribed by standards, in particular according to ISO/IEC 14496-3. In contrast to the present invention, in the mentioned working versions of the standards the values for the code books are present in unquantized form, in floating-point format, and can be processed directly only using very expensive and memory-intensive methods. Despite the limitation of precision of the table values, in the present invention an equal subjective quality is to be achieved after the speech decoding. Using the measures of the present invention, a simple transfer – conforming to standards – of the code to various computing platforms is possible without influencing the subjective quality of the coder. Since reduced word lengths are used, a considerable savings of memory capacity, in particular in the form of ROMs, is possible. The present invention can be used with various speech signal coding methods, for example for HVXC coders/decoders or CELP coders/decoders.

Brief Description Of The Drawings

[Exemplary embodiments of the invention are explained in more detail on the basis of the drawings.]

Figure 1 shows a simplified block switching diagram of an HVXC speech decoder[, and].

Figure 2 shows a simplified block switching diagram of a CELP speech decoder.

Detailed Description [of Exemplary Embodiments]

Before discussing the actual quantization, a speech decoder is first presented in which the inventive quantization is used.

In the HVXC speech decoder according to Figure 1, the transmitted speech parameters,

namely the LPC parameters, the voiced/unvoiced decision of the encoder, and the excitation parameters, which are contained in a transmission frame of 20 ms duration, are read out from the bitstream and are supplied as input signals to inputs 1, 2, and 3. The LPC parameters contain indices from which inverse LSP vector quantizer 16 regenerates the LSP (Line Spectral Pairs) parameters. For this purpose, LSP code books 4 (CbLsp) and 5 (CbLsp4) are indexed with the LPC parameters, and the LSP parameters are read out. Dependent on the voiced/unvoiced decision of this frame, if necessary interpolation – module 6 – takes place between the LSP parameters of the past and current frame, achieving an updating of these values in a raster of 2.5 ms. Subsequently, conversion takes place into LPC parameters, which enter as coefficients into the LPC synthesis filter – modules 7 and 8.

Parallel to this calculation, and as a function of the voiced/unvoiced decision, the vectors for the spectral envelope (voiced frame), AM code books 9 (CbAm) and 10 (CbAm4), or the vectors for the stochastic excitation signal (unvoiced frame, CELP code books 11 (CbCelp) and 12 (CbCelp4)) are read. The regeneration of the spectral envelopes and of the excitation signal takes place using the inverse vector quantizers 13 and 14. After the harmonic synthesis (voiced) – module 15 – the filtering of the speech data takes place in the LPC synthesis filter. The output data from the voiced - module 7 - and from the unvoiced - module 8 - synthesis filter are subsequently added, yielding the reconstructed speech signal for a frame of 20 ms.

Because, as explained above, values for the code books in floating-point form are not suitable for fixed-point DSPs, because the required word lengths would be too large (memory requirement, internal word lengths and arithmetic, ROM), the conversion of the table values for the code books that were previously obtained by analysis from the speech signal sampled values takes place in a quantized form, with resulting equivalent speech quality. The word lengths required for this for the individual table values are determined in various hearing tests.

The quantization takes place to a word length that is determined in various tests. In the following representation, this word length is designated in general as *wordlength*. This size is expressed in bits. A signed whole number having *wordlength* bits includes a value range from $-2^{wordlength-1}$ to $2^{wordlength-1}-1$. The quantization of the code books in this context takes place in

the manner shown below. The beginning point is represented by the code books defined in the "Study on ISO/IEC 14496-3 FCD, Subpart 3." For this document, the code book *cb* is defined as follows: $cb = \{a0, a1, \ , an, ..., am\}$ with $0 \leq n \leq m$ and $an \in R$. For the quantization of the individual elements, the following steps are required:

1.) Determination of the value range of the code books

In order to obtain a well-matched quantization, the elements of each code book are scaled in such a manner that the available value range is exploited as completely as possible. For this purpose, the value range of the elements [must be] is located between

$$\frac{-2^{wordlength-1}}{2^{wordlength-1}} = -1 \quad \textbf{and} \quad \frac{2^{wordlength-1}-1}{2^{wordlength-1}} = 1 - 2^{-(wordlength-1)}$$

In order to achieve this, the maximum of the positive and of the negative elements (*max_pos* or *max_neg*) of each code book [must be] is determined. These result from

$$max\_pos = \max \left(\left\{a_n \in cb \middle| a_n \geq 0\right\}\right) \text{ or } max\_neg = \min \left(\left\{a_n \in cb \middle| a_n \geq 0\right\}\right), \text{ with } 0 \leq n \leq m.$$

As a function of the magnitude of *max_pos* or *max_neg*, the following steps result:

$max\_pos > (1 - 2^{-(wordlength-1)})$ or $max\_neg < -1$

*max_pos* and *max_neg* are multiplied by ½. If the result still satisfies the condition set under (a), then the process [must be] is repeated until the condition no longer holds. The number of multiplications by ½ is counted and is stored in the variables *scale*.

$max\_pos \leq (1 - 2^{-(wordlength-1)})$ or $max\_neg \geq -1$

*max_pos* and *max_neg* are multiplied by 2. If the result still satisfies the condition set under (b), then the process [must be] is repeated until the condition no longer holds. The number of multiplications by 2 is counted and is stored in the variables *scale*.

2.) Scaling of the elements of *cb* to the range between -1 and $(1 - 2^{-(wordlength - 1)})$.

As a function of the decision made under 1.), the scaling of all code book entries to the cited range takes place:

$$b_n = \frac{1}{2^{scale}} a_n \forall a_n \in cb \text{ with } 0 \leq n \leq m$$

$$b_n = 2^{scale} a_n \forall a_n \in cb \text{ with } 0 \leq n \leq m.$$

After this step, the entries of each code book are located in the following range of values: $-1 \leq b_n \leq (1 - 2^{-(wordlength - 1)})$, with $0 \leq n \leq m$.

3.) Scaling to *wordlength* bits

For the scaling to the required value range, multiplication by $2^{wordlength - 1}$ takes place. In this way, the values of code books $c^n$ are located in the range between $-2^{wordlength - 1}$ and $2^{wordlength - 1} - 1$.

4.) Rounding

Before the decimal places are truncated, rounding of the determined entries takes place. For this purpose, depending on the sign + 0.5 or – 0.5 is added. This takes place in the following form:

$c_n \geq 0 : d_n = c_n + 0.5$

$c_n < 0 : d_n = c_n - 0.5$

Here care is to be taken not to exceed the maximum permissible value range. This is located in the range as indicated under 2.).

5.) Truncation of the decimal places

The final quantization takes place through the truncation of the decimal places. The quantized values are obtained in this way.

Trials have shown that with the setting of the variables *wordlength* at 16, a speech quality indistinguishable from the original is obtained.

A further construction of the present invention is explained in connection with [Fig.] Figure 2. There, the block switching diagram of a CELP decoder is shown. First, the elements [necessary] for decoding a frame are read from a transmitted bitstream, as before. These include the LPC indices, the excitation parameters (lag and shape index), and the amplitude indices (gain indices). These parameters (elements) are supplied to decoder inputs 17 to 21. The excitation parameters are made up of the parameters for adaptive code book (lag) 22 for the generation of periodic signal components (voiced) and the parameters for fixed code books (shape index) 23a ... 23n.

The entries of fixed code books 23a ... 23n and of adaptive code book 22 are each multiplied by a scaling factor (gain) via gain decoder 24. This scaling factor is reconstructed with the aid of the gain indices present at the input 21 and the gain VQ (vector quantization) tables stored in code books 25. The finally valid excitation vector is composed from the sum of the fixed and the adaptive code book vector.

With the use of vector quantizer VQ, the LPC indices represent the vector-quantized LSP (Line Spectral Pairs) parameters. The vectors of the first and second stage of the inverse vector quantization of the LSP parameters are obtained by reading out the LSP-VQ table values, which are stored in code books 26. The finally valid reconstruction of the LPC parameters takes place in LPC parameter decoder 27. Inside each frame, for each subframe interpolation - module 28 - takes place between the LSP parameters of the past and of the current frame. The LSP parameters, converted into LPC parameters, enter into LPC synthesis filter 29 as coefficients. The reconstruction of the speech data takes place there through filtering of the excitation signal. In order to improve the speech quality, the reconstructed speech signal can be additionally filtered in a post-filter 30.

The LSP VQ table values, as well as the gain VQ table values for code books 25 and 26, which were previously obtained by analysis from the speech signal sampled values, are normally present in a floating-point representation, which, as explained above, is not suitable

for a fixed-point DSP processing. For the same reasons as in the case of the HVXC decoder (Figure 1), a conversion of the table values into a quantized form takes place. The method steps in this quantization, such as in particular the determination of the value range for the code books, takes place as in the previously explained quantization.

The above exemplary embodiments of the present invention have been explained on the basis of speech decoders. Of course, the present invention can also be used in corresponding coders (encoders) that use code books. There as well, the code book entries can be previously quantized for the preparation of speech signals for transmission. Examples of such encoders whose code book entries can be previously quantized [are known from] described in European Published Patent Application No. 0545 386 [A2], U.S. Patent No. 5,208,862, U.S. Patent No. 5,487,128, U.S. Patent No. 5,199,076, or U.S. Patent No. 5,261,027.

Abstract Of The Disclosure

For the coding or decoding of speech signal sampled values, the values contained in the code books/code tables for the generation of the speech signal parameters are stored in quantized form. The processing can be carried out using processors with whole-number processing, without deterioration of the speech quality.

[Fig. 1]

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New Set of Claims

1. A method for coding or decoding speech signal sampled values, in particular using the method of analysis through synthesis, comprising the following steps:

- the values previously obtained by analysis from the speech signal sampled values and used for the generation of the speech signal parameters are quantized before being stored in code books/code tables,
- the quantization of the values takes place to a word length that results in no noticeable losses in speech quality, determined in particular through hearing tests,
- the values of each code book/code table are scaled such that the available range of values is exploited as completely as possible,

for the scaling, the maximum of the positive and negative values of each code book/code table is determined, and if the available value range is exceeded, a multiplication of the values by a factor smaller than one, preferably 0.5, takes place, and this multiplication is repeated until all elements are located in the value range,

- the number of repeated multiplications is used as a scaling factor for all code book/table entries.

2. The method according to Claim 1, wherein the word lengths of the values stored in the code books/code tables are determined through hearing tests.

3. The method according to Claim 1 or 2, wherein the values of each code book/code table are scaled such that the available range of values is exploited as completely as possible.

4. The method according to one of Claims 1 to 3, wherein a scaling of the code book/table entries to the bits of the required value range is carried out.

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5. The method according to Claim 4, wherein for a finally valid quantization, a rounding and a subsequent truncation of decimal places takes place.

6. The method according to Claims 1 to 5, wherein the word length of the quantized values is selected at 16 bits.

7. The method according to one of Claims 1 to 6, wherein the processing of the quantized code book/table entries takes place by means of digital signal processing, in whole-number format.

8. The method according to one of Claims 1 to 7, wherein for a HXVC (Harmonic Vector Excitation Coding) speech coder/speech decoder, the LPC coefficients, the spectral envelopes of the speech signal, and the unvoiced segments of the speech signal are stored in quantized form in the corresponding code books/tables.

9. The method according to one of Claims 1 to 5, wherein for a CELP (Code Excited Linear Prediction) speech coder/decoder, the values for the LSP (Line Spectral Pairs) VQ vector quantization **[sic]** code book/table entries, as well as those of the gain VQ table entries, are stored in quantized form.

10. A coder or decoder for processing speech signal sampled values using the method of analysis through synthesis, comprising the following measures:
- the values contained in the code books/code tables (4, 5, 9, 10, 11, 12, 25, 26) for the generation of the speech signal parameters are stored in quantized form, the word length being selected such that no noticeable losses in speech quality occur,
- means are provided for quantizing the values to a word

length that results in no noticeable losses in speech quality, determined in particular through hearing tests,

- means are provided for scaling the values of each code book/code table, such that the available range of values can be exploited as completely as possible,
- means are provided for determining the maximum of the positive and negative values of each code book/code table, and for multiplying these values by a factor less than one, preferably 0.5, if the available range of values is exceeded,
- for the case in which the multiplication of the latter values in turn lies outside the range of values, means are provided for repeated multiplication until all elements are located in the range of values, and for providing the number of repeated multiplications as a scaling factor.

# METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER

## Prior art

The invention is based on a method for coding or decoding speech signal sampled values.

In the standard for coding audiovisual objects according to MPEG-4, in ISO/IEC 14496-3 FCD, Subpart 2, parametric coders are specified, in particular the HVXC (Harmonic Vector Excitation Coding) coder, for coding speech at extremely low bitrates. In order to generate the LPC coefficients, the spectral envelopes of the speech signal, and the unvoiced segments, this standard contains a plurality of tables that are present in floating-point format.

In Subpart 3 of this standard, the CELP (Code Excited Linear Prediction) coder for coding speech at medium to low bitrates is described. For generating the LPC coefficients and the gain values, this standard contains a plurality of tables that are present in floating-point format.

For coding such speech signals, the method of "analysis through synthesis" is often used (ANT Nachrichtentechnische Berichte, Heft 5, Nov. 1988, pages 93 to 105). In the mentioned speech coding methods, values are stored in code books, i.e., in the tables, said values being used for the generation of the signal parameters and thus for the coefficients of the speech synthesis filter. The values stored in the code books are read out via an index control unit.

## Advantages of the Invention

Through the measures of Claim 1, i.e., in particular through the quantization of the values in the code books, the existing data are limited in their precision (quantization) so that the code book entries can be represented with a finite word length. In this way, their transfer to digital signal processors with whole-number arithmetic can take place without infringing the quality

demands prescribed by standards, in particular according to ISO/IEC 14496-3. In contrast to the present invention, in the mentioned working versions of the standards the values for the code books are present in unquantized form, in floating-point format, and can be processed directly only using very expensive and memory-intensive methods. Despite the limitation of precision of the table values, in the present invention an equal subjective quality is to be achieved after the speech decoding. Using the measures of the present invention, a simple transfer – conforming to standards – of the code to various computing platforms is possible without influencing the subjective quality of the coder. Since reduced word lengths are used, a considerable savings of memory capacity, in particular in the form of ROMs, is possible. The invention can be used with various speech signal coding methods, for example for HVXC coders/decoders or CELP coders/decoders.

Drawings

Exemplary embodiments of the invention are explained in more detail on the basis of the drawings.

Figure 1 shows a simplified block switching diagram of an HVXC speech decoder, and

Figure 2 shows a simplified block switching diagram of a CELP speech decoder.

Description of Exemplary Embodiments

Before discussing the actual quantization, a speech decoder is first presented in which the inventive quantization is used.

In the HVXC speech decoder according to Figure 1, the transmitted speech parameters, namely the LPC parameters, the voiced/unvoiced decision of the encoder, and the excitation parameters, which are contained in a transmission frame of 20 ms duration, are read out from the bitstream and are supplied as input signals to inputs 1, 2, and 3. The LPC parameters contain indices from which inverse LSP vector quantizer 16 regenerates the LSP (Line Spectral Pairs) parameters. For this purpose, LSP code books 4 (CbLsp) and 5 (CbLsp4) are indexed with the LPC parameters, and the LSP parameters are read out. Dependent on the voiced/unvoiced decision of this frame, if necessary interpolation – module 6 – takes place between the LSP parameters of the past and current frame, achieving an updating of these values in a raster of 2.5 ms. Subsequently, conversion takes place into LPC parameters,

which enter as coefficients into the LPC synthesis filter – modules 7 and 8.

Parallel to this calculation, and as a function of the voiced/unvoiced decision, the vectors for the spectral envelope (voiced frame), AM code books 9 (CbAm) and 10 (CbAm4), or the vectors for the stochastic excitation signal (unvoiced frame, CELP code books 11 (CbCelp) and 12 (CbCelp4)) are read. The regeneration of the spectral envelopes and of the excitation signal takes place using the inverse vector quantizers 13 and 14. After the harmonic synthesis (voiced) – module 15 – the filtering of the speech data takes place in the LPC synthesis filter. The output data from the voiced - module 7 - and from the unvoiced - module 8 - synthesis filter are subsequently added, yielding the reconstructed speech signal for a frame of 20 ms.

Because, as explained above, values for the code books in floating-point form are not suitable for fixed-point DSPs, because the required word lengths would be too large (memory requirement, internal word lengths and arithmetic, ROM), the conversion of the table values for the code books that were previously obtained by analysis from the speech signal sampled values takes place in a quantized form, with resulting equivalent speech quality. The word lengths required for this for the individual table values are determined in various hearing tests.

The quantization takes place to a word length that is determined in various tests. In the following representation, this word length is designated in general as *wordlength*. This size is expressed in bits. A signed whole number having *wordlength* bits includes a value range from $-2^{wordlength-1}$ to $2^{wordlength-1}-1$. The quantization of the code books in this context takes place in the manner shown below. The beginning point is represented by the code books defined in the "Study on ISO/IEC 14496-3 FCD, Subpart 3." For this document, the code book *cb* is defined as follows: $cb = \{a0, a1, .., an, .., am\}$ with $0 \le n \le m$ and $an \in R$. For the quantization of the individual elements, the following steps are required:

1.) Determination of the value range of the code books

In order to obtain a well-matched quantization, the elements of each code book are scaled in such a manner that the available value range is exploited as completely as possible. For this purpose, the value range of the elements must be located between

$$\frac{-2^{wordlength-1}}{2^{wordlength-1}} = -1 \quad \textbf{and} \quad \frac{2^{wordlength-1}-1}{2^{wordlength-1}} = 1 - 2^{-(wordlength-1)}$$

In order to achieve this, the maximum of the positive and of the negative elements (*max_pos* or *max_neg*) of each code book must be determined. These result from

$$max\_pos = \max\left(\left\{a_n \in cb \middle| a_n \geq 0\right\}\right) \text{ or } max\_neg = \min\left(\left\{a_n \in cb \middle| a_n \geq 0\right\}\right) \text{, with } 0 \leq n \leq m.$$

As a function of the magnitude of *max_pos* or *max_neg*, the following steps result:

$max\_pos > (1 - 2^{-(wordlength-1)})$ or $max\_neg < -1$

*max_pos* and *max_neg* are multiplied by ½. If the result still satisfies the condition set under (a), then the process must be repeated until the condition no longer holds. The number of multiplications by ½ is counted and is stored in the variables *scale*.

$max\_pos \leq (1 - 2^{-(wordlength-1)})$ or $max\_neg \geq -1$

*max_pos* and *max_neg* are multiplied by 2. If the result still satisfies the condition set under (b), then the process must be repeated until the condition no longer holds. The number of multiplications by 2 is counted and is stored in the variables *scale*.

2.) Scaling of the elements of *cb* to the range between -1 and $(1 - 2^{-(wordlength-1)})$.

As a function of the decision made under 1.), the scaling of all code book entries to the cited range takes place:

$$b_n = \frac{1}{2^{scale}} a_n \forall a_n \in cb \text{ with } 0 \leq n \leq m$$

$$b_n = 2^{scale} a_n \forall a_n \in cb \text{ with } 0 \leq n \leq m.$$

After this step, the entries of each code book are located in the following range of values: $-1 \leq b_n \leq (1 - 2^{-(wordlength-1)})$, with $0 \leq n \leq m$.

3.) Scaling to *wordlength* bits

For the scaling to the required value range, multiplication by $2^{wordlength-1}$ takes place. In this way, the values of code books $c''$ are located in the range between $-2^{wordlength-1}$ and $2^{wordlength-1}-1$.

4.) Rounding

Before the decimal places are truncated, rounding of the determined entries takes place. For this purpose, depending on the sign + 0.5 or – 0.5 is added. This takes place in the following form:

$c_n \geq 0 : d_n = c_n + 0.5$

$c_n < 0 : d_n = c_n - 0.5$

Here care is to be taken not to exceed the maximum permissible value range. This is located in the range as indicated under 2.).

5.) Truncation of the decimal places

The final quantization takes place through the truncation of the decimal places. The quantized values are obtained in this way.

Trials have shown that with the setting of the variables *wordlength* at 16, a speech quality indistinguishable from the original is obtained.

A further construction of the invention is explained in connection with Fig. 2. There, the block switching diagram of a CELP decoder is shown. First, the elements necessary for decoding a frame are read from a transmitted bitstream, as before. These include the LPC indices, the excitation parameters (lag and shape index), and the amplitude indices (gain indices). These parameters (elements) are supplied to decoder inputs 17 to 21. The excitation parameters are made up of the parameters for adaptive code book (lag) 22 for the generation of periodic signal components (voiced) and the parameters for fixed code books (shape index) 23a ... 23n.

The entries of fixed code books 23a ... 23n and of adaptive code book 22 are each multiplied

by a scaling factor (gain) via gain decoder 24. This scaling factor is reconstructed with the aid of the gain indices present at the input 21 and the gain VQ (vector quantization) tables stored in code books 25. The finally valid excitation vector is composed from the sum of the fixed and the adaptive code book vector.

With the use of vector quantizer VQ, the LPC indices represent the vector-quantized LSP (Line Spectral Pairs) parameters. The vectors of the first and second stage of the inverse vector quantization of the LSP parameters are obtained by reading out the LSP-VQ table values, which are stored in code books 26. The finally valid reconstruction of the LPC parameters takes place in LPC parameter decoder 27. Inside each frame, for each subframe interpolation - module 28 - takes place between the LSP parameters of the past and of the current frame. The LSP parameters, converted into LPC parameters, enter into LPC synthesis filter 29 as coefficients. The reconstruction of the speech data takes place there through filtering of the excitation signal. In order to improve the speech quality, the reconstructed speech signal can be additionally filtered in a post-filter 30.

The LSP VQ table values, as well as the gain VQ table values for code books 25 and 26, which were previously obtained by analysis from the speech signal sampled values, are normally present in a floating-point representation, which, as explained above, is not suitable for a fixed-point DSP processing. For the same reasons as in the case of the HVXC decoder (Figure 1), a conversion of the table values into a quantized form takes place. The method steps in this quantization, such as in particular the determination of the value range for the code books, takes place as in the previously explained quantization.

The above exemplary embodiments of the present invention have been explained on the basis of speech decoders. Of course, the invention can also be used in corresponding coders (encoders) that use code books. There as well, the code book entries can be previously quantized for the preparation of speech signals for transmission. Examples of such encoders whose code book entries can be previously quantized are known from European Patent 0545 386 A2, U.S. Patent 5,208,862, U.S. Patent 5,487,128, U.S. Patent 5,199,076, or U.S. Patent 5,261,027.

Patent Claims

1. A method for coding or decoding speech signal sampled values, in particular using the method of analysis through synthesis, comprising the following steps:

– the values previously obtained by analysis from the speech signal sampled values and used for the generation of the speech signal parameters are quantized before being stored in code books/code tables,

– the quantization of the values takes place to a word length that results in no noticeable losses in speech quality.

2. The method according to Claim 1,
wherein the word lengths of the values stored in the code books/code tables are determined through hearing tests.

3. The method according to Claim 1 or 2,
wherein the values of each code book/code table are scaled such that the available range of values is exploited as completely as possible.

4. The method according to Claim 3,
wherein for the scaling, the maximum of the positive and negative values of each code book/code table is determined,
and if the available value range is exceeded, a multiplication of the values by a factor smaller than one, preferably 0.5, takes place,
and this multiplication is repeated until all elements are located in the value range.

5. The method according to Claim 4,
wherein the number of repeated multiplications is used as a scaling factor for all code book/table entries.

6. The method according to Claim 5,
wherein a scaling of the code book/table entries to the bits of the required value range is carried out.

7. The method according to Claim 6,
wherein for a finally valid quantization, a rounding and a subsequent truncation of decimal places takes place.

8. The method according to one of Claims 1 to 7,
wherein the word length of the quantized values is selected at 16 bits.

9. The method according to one of Claims 1 to 8,
wherein the processing of the quantized code book/table entries takes place using digital signal processing, in whole-number format.

10. The method according to one of Claims 1 to 9,
wherein for a HXVC (Harmonic Vector Excitation Coding) speech coder/speech decoder, the LPC coefficients, the spectral envelopes of the speech signal, and the unvoiced segments of the speech signal are stored in quantized form in the corresponding code books/tables.

11. The method according to one of Claims 1 to 7,
wherein for a CELP (Code Excited Linear Prediction) speech coder/decoder, the values for the LSP (Line Spectral Pairs) VQ vector quantization code book/table entries, as well as those of the gain VQ table entries, are stored in quantized form.

12. A coder or decoder for processing speech signal sampled values using the method of analysis through synthesis, comprising the following measures:
the values contained in the code books/code tables (4, 5, 9, 10, 11, 12, 25, 26) for the generation of the speech signal parameters are stored in quantized form, the word length being selected such that no noticeable losses in speech quality occur.

Fig. 1

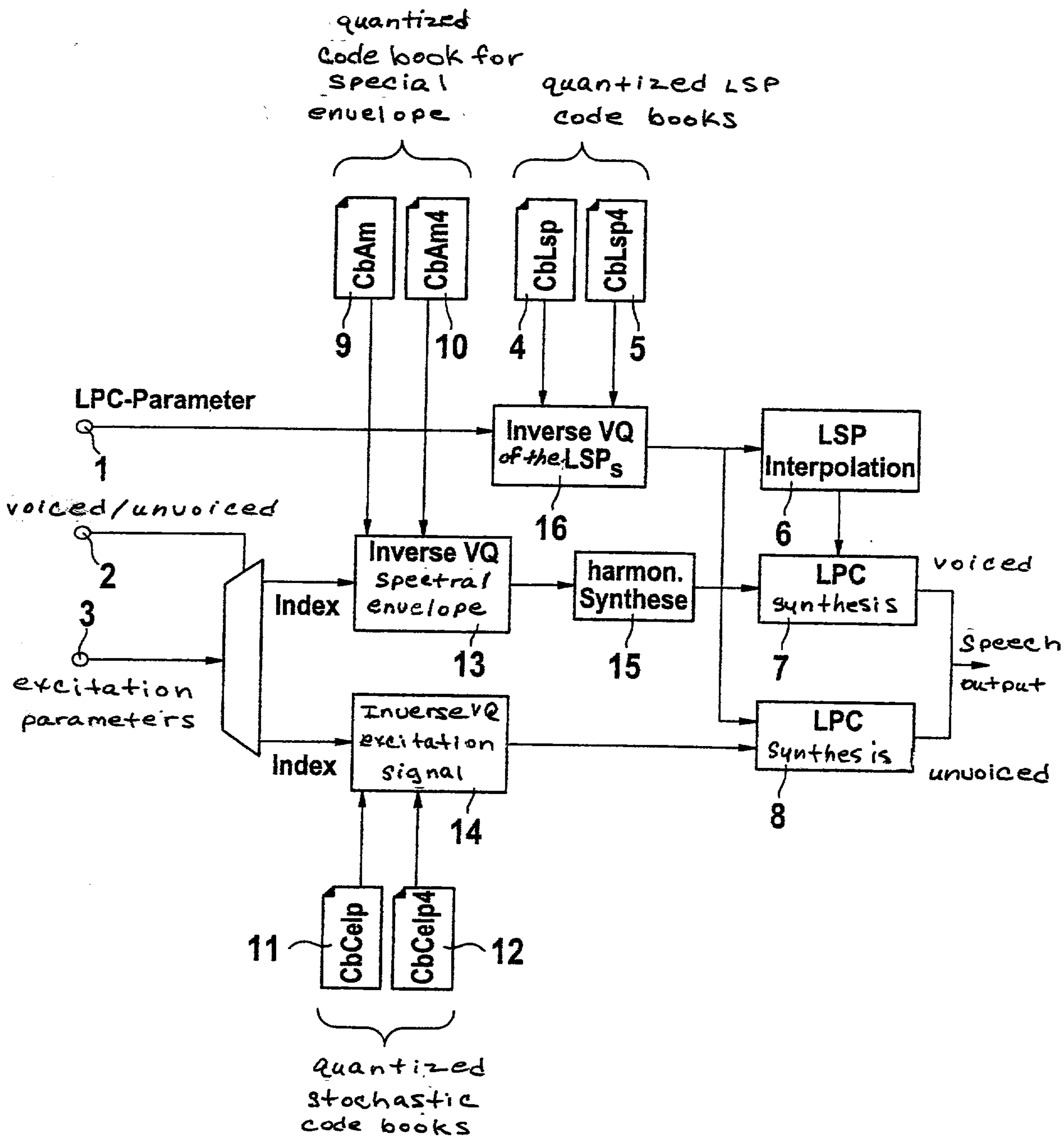
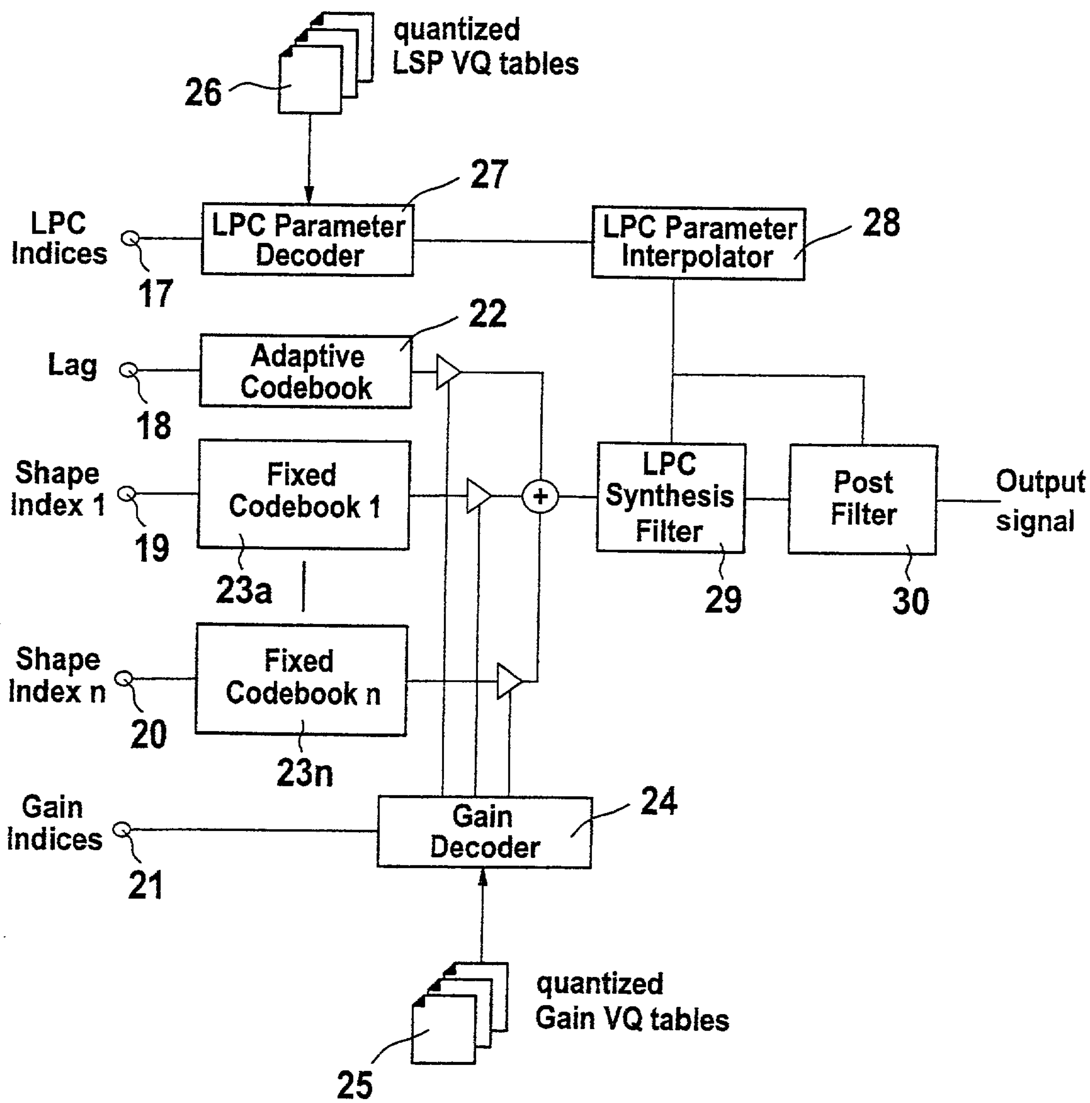

quantized
code book for
special
envelope
quantized LSP
code books
CbAm
CbAm4
CbLsp
CbLsp4
9
10
4
5
LPC-Parameter
1
Inverse VQ
of the LSPs
16
LSP
Interpolation
6
voiced/unvoiced
2
3
excitation
parameters
Index
Inverse VQ
spectral
envelope
13
harmon.
Synthese
15
LPC
synthesis
7
voiced
Speech
output
Index
Inverse VQ
excitation
signal
14
LPC
Synthesis
8
unvoiced
11
CbCelp
CbCelp4
12
quantized
stochastic
code books

Fig. 2


quantized
LSP VQ tables
26
LPC Parameter Decoder
27
LPC Parameter Interpolator
28
LPC Indices
17
Adaptive Codebook
22
Lag
18
Shape Index 1
19
Fixed Codebook 1
23a
LPC Synthesis Filter
29
Post Filter
30
Output signal
Shape Index n
20
Fixed Codebook n
23n
Gain Indices
21
Gain Decoder
24
quantized
Gain VQ tables
25

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[10191/1822]

**DECLARATION AND POWER OF ATTORNEY**

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe I am an original, first and joint inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled **METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER**, the specification of which was filed as International Application No. PCT/DE99/02633 on August 21, 1999.

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, § 1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, § 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application(s) for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

## PRIOR FOREIGN APPLICATION(S)

| Number | Country Filed | Day/Month/Year | Priority Claimed Under 35 USC 119 |
|---|---|---|---|
| 198 45 888.6 | Fed. Rep. of Germany | 6 October 1998 | Yes |

And I hereby appoint Richard L. Mayer (Reg. No. 22,490) and Gerard A. Messina (Reg. No. 35,952) my attorneys with full power of substitution and revocation, to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith.

Please address all communications regarding this application to:

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26646
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Please direct all telephone calls to Richard L. Mayer at (212) 425-7200.

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful and false statements may jeopardize the validity of the application or any patent issued thereon.

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